REMARKS

Status of Claims:

The examiner is reminded that the present Reply is **supplemental** to the previously filed amendment mailed on December 21, 2004.

New claims 30-44 have been added. Thus, claims 10-18 and 20-44 are present for examination.

Prior Art Rejection:

Claims 10, 11 and 14-18 and 20 stand rejected under 35 U.S.C. § 103 as obvious over Nomura (CA 2,112,145) in view of applicant's admitted prior art APA. Further, claims 12 and 13 stand rejected under 35 U.S.C. § 103(a) as unpatentable over Nomura in view of APA and further in view of Takada (US 6,088,670). The Examiner's rejections are respectfully traversed.

Applicant's claim 10 recites:

> 10. (Currently Amended) A speech signal decoding apparatus comprising:
>
> a plurality of decoding means for decoding information containing at least a sound source signal, a gain, and filter coefficients from a received bit stream;
>
> identification means for identifying voiced speech and unvoiced speech of a speech signal using the decoded information, <u>at least the unvoiced speech containing a background noise;</u>
>
> smoothing means for performing smoothing processing based on the decoded information for at least either one of the decoded gain and the decoded filter coefficients in the speech identified by said identification means <u>in order to provide enhanced coding quality for at least the unvoiced speech with the background noise;</u>
>
> means for obtaining an excitation signal by multiplying the decoded sound source signal by the decoded gain after performing the smoothing processing; and

> means for decoding the speech signal by driving a filter having the decoded filter coefficients by the excitation signal obtained from the means for obtaining.

The primary prior art reference to Nomura neither discloses nor suggest the use of a smoothing processing means or device for enhancing the coding quality for the speech signal containing a background noise. The underlined portions of claim 10 set forth above emphasize this distinguishing limitation of applicant's claim. Moreover, similar limitations have been made to all of applicant's independent claims, including the newly added independent claims.

As such, it is submitted that all of applicant's claims are clearly patentable over the prior art, and that the PTO has not made out a *prima facie* case of obviousness under the provisions of 35 U.S.C. § 103.

With regard to new claims 33-41, applicant points out that these claims are written in non-means plus function form in order to take them outside of the statutory provisions of 35 U.S.C. §112 ¶6. These claims are otherwise similar to claims 10-18. Further, claim 42 is the non-means plus function counterpart to claim 30; claim 43 is the non-means plus function counterpart to claim 20; and claim 44 is the non-means plus function counterpart to claim 31.

The discussion set forth below further demonstrates the non-obviousness of applicant's recited limitations. These arguments were previously submitted in applicant's prior Reply and are repeated here for completeness.

The stated in applicant's prior response, the differences between applicant's invention and the primary Nomura reference may be seen from the below discussion.

The constitution according to (CA 2,112,145) (Nomura et al.) is as follows:

(1) As for the frame with errors in the speech parameters such as a gain and a spectral parameter (filter coefficients),

(2) by repeatedly using the speech data in the past frame by "bad frame masking unit",

(3) a speech signal is reproduced for the frame with errors.

In addition, although the "bad frame masking unit" is divided into the voiced one and the unvoiced one, each is considered to be the fixed processing.

On the other hand, the features according to the present invention are:

(a) in the section of unvoiced speech (a section of only the background noise without the speech),

(b) by using a smoothing method based on "decoded information" (the speech parameters such as a gain and filter coefficients indicating the background noise),

(c) smoothing of the speech parameters indicating the background noise is made in terms of time.

This feature is equivalent to changing the filter used for smoothing according to the characteristic of a background noise signal in an embodiment.

The object of the reference is to reproduce a speech signal for the frame with errors (to apply the above-mentioned processing to only the frame with errors), and it is completely different from the present invention which aims at an improvement of the sound quality of the background noise.

From the viewpoint of a constitution, according to the reference, the signal of the present frame is reproduced by repeatedly using the speech parameters of the past frame. For this reason, when it is applied to the transition section of the voiced speech and the unvoiced speech, or to the background noise in which power fluctuates in terms of time (for example, the noise in the office environment with various kinds of sound sources such as people's voices or call sounds of telephones, etc.). there is a problem of degrading the sound quality.

On the other hand, according to the present invention, the smoothing method based on the "decoded information" is used. Since the speech parameters of the "decoded information", that is, the gain and the filter coefficients, etc. indicate the character of the

background noise, it becomes possible, according to the present invention, to make the appropriate smoothing of said speech parameters by the smoothing method corresponding to the characteristic of the background noise. As a result, it has the effect that good speech quality can be obtained for said transition section and various kinds of noises also.

With regard to Takada, the Examiner has pointed out that the voice detector (col. 8, lines 13-53) according to Takada (US 6,088,670) corresponds to wherein said identification/classification means performs identification/classification operation using a value obtained by averaging for a long term a variation amount based on a difference between the decoded filter coefficients and their long-tent average" in claims 12, 13. However, the voice detector according to Takada inputs only a speech signal itself. Therefore, an application of Takada to the decoder which cannot use a speech signal as an input is impossible.

On the other hand, according to claims 12 and 13, the voice detection is performed based on the "decoded information (obtained from the input to an encoder)", such as "decoded filter coefficients". Therefore, claims 12 and 13, relating to a decoder and having the above-mentioned distinctive feature of using information as obtained from a decoder, have nothing to do with Takada.

In summary, Nomura describes a system for using a speech decoder for reducing the sound which is different from normal voiced speech that is generated in decoding of the frame which has an error in a speech parameter such as a gain or a spectral parameter (filter coefficient). Therefore, according to the decoding processing by Nomura, the reduction processing of the sound which is different from normal voiced speech that is generated is performed only for a frame which has an error.

On the other hand, the present invention reduces the sound which is different from normal voiced speech that is generated at the time of decoding due to the background noise signal included in the speech to be coded. Therefore, the decoding processing according to the present invention is performed at least in the unvoiced speech section regardless of an error of a speech parameter. On the other hand, according to Nomura, as long as there is no

error in the speech parameter in the unvoiced speech section, the sound different from normal voiced speech that is generated resulting from the background noise in the unvoiced speech section is not improved.

According to the present invention, there is an advantage, which cannot be expected in Nomura, that the decoding quality is improved on the whole, since the sound different from normal voiced speech that is generated resulting from the background noise in the unvoiced speech section is reduced.

In view of the above, the present invention, in comparison with the reference, is completely different in all respects of the object, the constitution, and the effect.

Conclusions:

The application is believed to be in condition for allowance and an early indication of same is earnestly solicited.

The Commissioner is hereby authorized to charge any additional fees which may be required regarding this application under 37 C.F.R. §§ 1.16-1.17, or credit any overpayment, to Deposit Account No. 19-0741. Should no proper payment be enclosed herewith, as by a check being in the wrong amount, unsigned, post-dated, otherwise improper or informal or even entirely missing, the Commissioner is authorized to charge the unpaid amount to Deposit Account No. 19-0741. If any extensions of time are needed for timely acceptance of papers submitted herewith, Applicant hereby petitions for such extension under 37 C.F.R. §1.136 and authorizes payment of any such extensions fees to Deposit Account No. 19-0741.

Respectfully submitted,

Date 1-14-05

By [signature]

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